<u>IN THE SPECIFICATION:</u>

Replace the paragraph beginning at Page 1, line 20, with the following paragraph:

SIP or Session Initiation Protocol is gaining ground both within enterprise networks and service provider networks as the vehicle for providing Voice Over IP (VoIP) and new multimedia services. Most VoIP applications today are stand-alone applications such as an IP Softphone, which runs on a user laptop allowing a user to place and receive voice calls. In addition, SIP is also in use for supporting Instant Messages between a client machine and a server machine. However, the SIP support is bundled with individual applications such an IM client or an IP softphone, and individual applications must run their own SIP support. This may create problems when multiple SIP protocol stacks are operational, e.g. SIP often uses port 5060 for its operation. If the bundled SIP stack within an application uses this port, it would mean that no other SIP application will be able to run on this port. More importantly, this is wasteful of system resources.

Replace the paragraph beginning at Page 4, line 16, with the following paragraph:

FIG. 5 is a flow diagram showing a method for implementing SIP service for applications that employ their own protocol definitions; and

Replace the paragraph beginning at Page 5, line 12, with the following paragraph:

SIP may be employed for enabling VoIP in enterprise and carrier networks. Desktop applications such as SAMETIME® (Instant Messaging) and email clients (LOTUS NOTES®), or web-browsers (such as Internet Explorer® (IE) or Netscape®) were either not enabled with voice or were decoupled from voice applications. However, in accordance with the present invention, with SIP-based VoIP, which is a text-based HTTP like protocol, it now becomes possible to at least (a) add another mode of user interaction/control to standard applications, e.g., voice (b) integrate the functionality of traditional telco-style applications such as teleconferencing, very easily into applications using SIP, (instead of using a separate mechanism to invoke such services) and(c) give rise to newer functionality arising out of tighter coupling between data and voice based applications, e.g., on-demand conferencing from applications.

Replace the paragraph beginning at Page 9, line 14, with the following paragraph:

A SIP API 46 permits different applications 41, 43, 45, and 47 to access the SIP service, e.g., web-browser, IM agent, email client, teleconference, etc. API 46 may transmit/receive media packets to/from the client machine 32, e.g., RTP voice packets, and provide~~s~~ a programmatic interface that permits other applications on the client machine 32 to invoke any of the above functionality (see also, e.g., FIGS. 2).

Replace the paragraph beginning at Page 9, line 22, with the following paragraph:

The SIP Service is designed to run as a system service, taking care of the VoIP communications for that machine. The Service is started at boot-time and it initiates a user-selected softphone 31 to enable receiving VoIP calls on the machine directly. Along with that it waits for any user commands (from its APIs) to perform desired functions. As an example, the user may click on an SIP URL in an INTERNET EXPLORER® window, which will make our protocol handler write on the SIP service socket a message indicating ~~it~~ to place the desired call. Internally, the SIP Service may include the following sub units:

Replace the paragraph beginning at Page 13, line 20, with the following paragraph:

The illustrative function provided in accordance with one embodiment of the present invention will now be described. In block 102, call creation is permitted in a desktop environment. A click of a link would place a call to an IP or PSTN telephone external to client 32. Similarly, calls can be terminated in block 104. Call transfer 106 may be performed by clicking a hypertext link or an icon for a particular person or machine. A subscription or notification message may be set up in block 108. This may include, for example, indicating to the system 30 that client machine 32 be alerted to a status of another telephone system or SIP based VoIP system. In other words, is that person or machine available for a phone call.

Replace the paragraph beginning at Page 15, line 14, with the following paragraph:

In another embodiment, Web click-to-call service may be provided. Web browsers ~~who~~ are made to recognize SIP URLs (e.g., by adding an entry in an operating system registry by associating a protocol handler for SIP). When a web page includes a SIP URI, this is recognized by the browser and shown as a 'clickable' link to the user. Clicking on this link causes the soft phone 34 to place a VoIP call to the URI, which may be

for a point-to-point call or a multi-party conference.

Replace the paragraph beginning at Page 21, line 10, with the following paragraph:

For applications that do not consult the common registry (e.g., LOTUS NOTES® and other LOTUS® products) and also for operating systems that do not have such a registry mechanism, custom hook-ups are provided for all the applications that ~~a~~ need to be enabled with VoIP. In such a case, three operations are preferably performed:

Replace the paragraph beginning at Page 27, line 11, with the following paragraph:

Alternately, step 410 can be bypassed ~~in~~ if the phone~~s~~ numbers of the parties are already known or designated. Another example of using an IM client is also described~~ further~~.